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(54) **Receiving equipment for the recognition of a number of different and predetermined frequency combinations of voice-frequency signalling components under the protection of a spurious-signal/voice-signal guard function.**

(57) Universal signalling receiver containing a speech detector specially developed for tone push-button signalling. To this end two side lobes are added to each detector, operating as DFT detectors with a main lobe for a relevant frequency of the eight nominal tone push-button signalling frequencies. This is achieved by making use in the DFT calculations of special product coefficients, which have been previously calculated for the low and high bands, respectively, of the four nominal signalling frequencies and stored in memories (9, 10), for particular Kaiser-Bessel window functions and for pairs of frequencies of 603 and 1039 Hz added pairwise to the low-band and 1107 and 1805 Hz added pairwise to the high-band. These coefficients are chosen such that the maxima of these added side lobes are $1/\sqrt{3}$-part of the maxima of the relevant main lobes. The speech detector comprises a function section 12 which adds together the correlator result other than the two strongest results as generated per cycle of 8 DFT's, and a comparator section 11 in which such a sum result is repeatedly compared with a fixed noise threshold. By means of such a speech detector the bandwidth involved in speech guarding is increased at the upper and lower sides, while the selectivity in these bands is improved.

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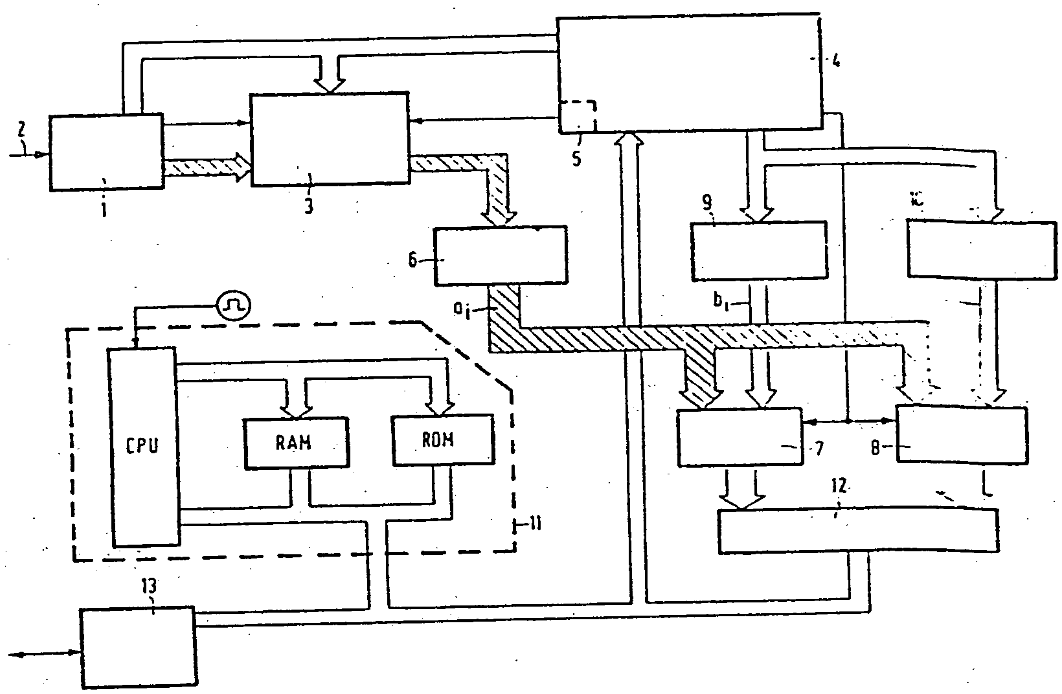
FIG.1



"Receiving equipment for the recognition of a number of different and predetermined frequency combinations of voice-frequency signalling components under the protection of a spurious-signal/voice-signal guard function."

The invention relates to a receiving equipment which is arranged to recognize out of incoming signals a number of different frequency combinations, each composed of at least two different combinations of a number (s) of nominal voice-frequency signalling components. To this end such a receiving equipment contains a number (s) of mutually different DTF digital signal-processing devices each having a frequency response characteristic the top of whose main lobe corresponds to one of the said nominal voice-frequency signalling components situated within a frequency band as determined by one of two subgroups into which said voice-frequency signalling components are divided; and a processor which is equipped to process the results of the digital signal processing devices, to detect two nominal voice-frequency signalling components that are received with the greatest strength, and to evaluate the other results so as to fulfil a voice-signal guard function.

Such a receiving equipment is known from Netherlands patent application PHN 10.240. This application describes an equipment for receiving multifrequency code signalling provided with DFT signal processing devices in which provisions are made to secure immunity against single spurious signal frequencies. These signal-guard provisions are not suitable for voice-signal protection and are further equipped in such a way as to reduce to a minimum the extra computing time required.

In the United States patent 4,355,405 a receiver for push-button signalling is described. Such a receiver comprises eight different digital signal-processing devices in the form of digital filters for the eight voice-frequency signalling components that are customary in such a sig-

nalling system. Such filters are implemented in this case in the form of "Finite Impulse Response" filters whose results are used on the one hand for detecting the relevant frequency combinations of signalling components, and on the other hand for fulfilling a voice-signal guard function. In this prior art it is proposed to add a side lobe to each filter transfer characteristic. The frequencies corresponding to the tops of these side lobes are chosen in such a way that one side-lobe top frequency is added concurrently to the four pairs of nominal signalling frequencies, i.e. the pairs 697, 770; 852, 941; 1209, 1336; and 1477, 1633, namely 1040 for the first pair, 640 for the second pair, 580 for the third pair and 1100 for the fourth pair. A filter system implemented in this way has the drawback that there are gaps in the voice-frequency spectrum which it analyses. This means that certain voice or spurious signal components, or a combination of nominal signalling components can produce imitations or cause the rejection of an incoming validating combination of nominal signalling components. Furthermore the selectivity of the filter characteristics thus divided over the frequency range is unsatisfactory. Nor does the literature referred to deal with provisions that would provide immunity in such pushbutton signalling against single spurious signal frequencies.

The object of the invention is to meet the objections to the prior art according to the above-mentioned Netherlands patent application by providing a receiving equipment for pushbutton signalling which has an effective voice-signal guard that is superior to the one known from the USA patent 4,355,405.

To this end the receiving equipment according to the invention is characterized in that the frequency response characteristic of each of the DFT digital signal-processing devices has a first side lobe and a second side lobe, each with one top, wherein the top of the first side lobe and the top of this second side lobe are situated

respectively below and above the frequency band of the subgroup to which the relevant nominal signalling frequency component belongs; and the processor is equipped a) for repeatedly adding together the results of a number (s-2) of the DFT devices in such a way that a sum result is obtained that is representative of the total energy content of all detected input signals other than two nominal voice-frequency signalling components that are received with the greatest strength, and b) for comparing this sum result with a threshold value which is derived from one of said two nominal voice-frequency signalling components that are received with the greatest strength. Such a receiving equipment designed for touch-tone signalling according to the invention in fact determines for the spurious-signal/voice-signal guard function the signal-to-noise ratio between a detected frequency combination of two nominal signalling components and the signal that corresponds to the energy content of the six other detectors, whereby this signal-to-noise ratio is compared with the derived threshold signal. When it appears from such a comparison that this signal-to-noice ratio is too low, the detected frequency combination of the relevant two signalling components is rejected.

In a spurious-signal/voice-signal guard device implemented in this manner the effectiveness of the guard function is in particular dependent on the size of the frequency band analysed. This means that the frequency response characteristic of each of the DFT devices must be as broad as possible. On the other hand such a frequency response characteristic should introduce a sufficiently high attenuation for the neighbouring nominal frequency of the same subgroup. In connection with these considerations a receiving equipment according to the invention is further characterized in that each of the digital signal processing devices is dimensioned in such a way that at the minimum frequency distance between a relevant nominal signalling frequency component and a frequency that limits

the detection bandwidth of the nearest neighbouring nominal signalling frequency component an attenuation is introduced which in the main has one and the same magnitude.

With such a device all detectors that are effective for the nominal voice-frequency signalling components of the subgroup covering the lowest frequency band are in fact effective for introducing at the minimum frequency distance to the relevant neighbouring frequency an identical attenuation value, as a result of which the bandwidth of these detectors is as broad as possible.

Further a receiving equipment according to the invention is characterized in that each of the digital signal processing devices is dimensioned such that the tops of the relevant two side lobes are mainly 6 dB lower than the top of the corresponding main lobe.

With such an embodiment the amplitude of a relevant side lobe is in fact given as $1/\sqrt{3}$ times the amplitude of the corresponding main lobe. In this way it is achieved that in each frequency band the level of the signals obtained from the added side lobes is equal to that of the signals of the main lobes.

In order in an effective manner to meet the imposed requirements, a receiving equipment according to the invention is further characterized in that the shape of the frequency response characteristic of each of the digital processing devices is determined by a window function of the kind known as the Kaiser-Bessel function.

It is remarked that such window functions and their application in combination with discrete Fourier transforms are generally known, for example from an article by F.J. Harris "On the use of windows for harmonic analysis with the discrete Fourier transformation", published in Proceedings of the IEEE, Vol. 66, No. 1, January 1978.

A preferred embodiment of a receiving equipment according to the invention is further characterized in

that the digital signal processing devices are so dimensioned that the frequencies of the tops of the side lobes that are added to the subgroup with the lowest and highest frequency band, respectively, are located at respectively 603 and 1039 Hz and 1107 and 1805 Hz.

It has been found that for touch-tone signalling detection the recommended number of samples of the signal to be analysed is N=256. It has also been found that for detection in an MFC system it is recommendable to use N=128 samples. A receiving equipment as known from aforesaid Netherlands patent application is provided with an input memory device for the temporary storage of a number of samples of a presented signal. In connection with the desired compatibility, a receiving equipment according to the invention is further characterized in that read-out means are provided for alternately reading out half the number of samples stored in the memory device, provided that upon each read-out the most neighbouring sample of a read-out sample is negated.

With such a device the sampling frequency is in fact halved from e.g. 8000 Hz to 4000 Hz. In this way the shape of a frequency response characteristic is not affected, although the frequency response in connection with the spurious signal/voice signal guard function, is favourable since it has in fact the effect of doubling the frequency band analysed for voice signals.

The invention will be explained in more detail in the following, together with an example of an embodiment thereof, with reference to the drawings in which:

fig. 1 shows a greatly simplified block diagram of a physical component structure illustrating an exemplary embodiment of a receiving equipment according to the invention;

fig. 2 shows a family of frequency response characteristics illustrating DFT devices usable for touch-tone signalling detection;

figs. 3 and 4 show families of frequency response

characteristics to illustrate parts of a DFT system such as used in a receiving equipment according to the invention; and

fig. 5 shows a graph giving the relation between the parameter a of a Kaiser-Bessel window function and the width of the main lobe, with corresponding attenuations for a neighbouring frequency.

The design of the example of an embodiment of a receiving equipment according to the invention, shown schematically in fig. 1, is based on the following considerations:

1. For detection of the nominal voice-frequency signalling components a discrete Fourier transformation (DFT) is carried out for each of the eight nominal signalling frequencies which are usual in a tone push-button signalling system (TDK system).
2. Use is also made of DFT for fulfilling the spurious-signal/voice-signal guard function.
3. Kaiser-Bessel window functions are used in the implantation of the various DFT's.
4. The various DFT's are implemented in overlapping form.
5. A digital signal processing device for a physical unit is available with which the real and the imaginary part of a DFT can be calculated over a maximum number of 128 samples in a frame time interval of normally 125 μs. Such a digital signal processing device operates in essence as a numerical series multiplier for implementing the operation:

$$S = \sum_{i=0}^{N-1} a(i) * b(i)$$

where a(i) is the signal input sample and b(i) and c(i) are quantities yet to be described, a(i) and b(i) being respectively a word (number) with a width of 16 and 8 bits respectively. For the sum S a word width of 24 bits is available.

For the implementation of each 8 DFT's a time

interval of 8 x 125 /us = 1 ms is therefore needed, using 8 digital signal processing devices of this physical unit. This offers the practical advantage of having one digital signal processing device for determining the DFT's of the eight nominal frequencies and of using these eight times in time division multiplex. It is further of advantage to make common use of the detection part, as shown schematically in figure 1, for eight channels stacked in time-division multiplex. In this way new information about the input signal can be obtained every eight milliseconds. As will presently be explained in more detail, the implementation of a DFT for each of the frequency/channel combinations requires a number of 256 samples. Since each of these samples appears after every time interval of 125 /us, this implies an acquisition interval of 256 x 125 /us = 32 ms. By overlapping the detection processes of the eight time-division multiplex stacked channels, this means that for each DFT an "old" signal segment corresponding to an acquisition interval of 24 ms is involved and a "new" signal segment corresponding to an acquisition interval of 8 ms.

The receiving equipment shown schematically in figure 1 can be used in common for a number of eight channels stacked in time-division multiplex. Signalling can take place through each of these channels with the codes customarily used for touch-tone signalling. These codes are composed of series of two voice-frequency signalling components, being a component from a lower group containing the frequencies 697; 770; 852; and 941, and a component from an upper group containing the frequencies 1209; 1336; 1477; and 1633. The block indicated by 1 is a data channel matching unit via which the data stream entering at input 2, which stream coincides the eight channels stacked in time-division multiplex, is matched to the input of the receiver. It is assumed that these signals are available in each channel as pulse-code modulated (PCM) signals. The signalling receiver shown

in fig. 1 comprises in essence eight digital signal processing devices each with a frequency response characteristic wherein a main lobe has its maximum or top centred at the relevant voice-frequency signalling component to be detected, which are referred to hereinafter as the nominal signalling frequencies. Fig. 2 shows the situation of this set of main lobes of the frequency response characteristics of these DFT devices for the frequency scheme normally used for touch-tone signalling with the two subgroups defined in the foregoing, each with four nominal signalling frequencies. There is thus a low-band subgroup and a high-band subgroup. Such a digital signal processing device operates on a series of numbers presented at its input, each of which represents a sample of the signal to be analysed, and uses a discrete Fourier transform to transform this series into a frequency response of said series of input signals. In a signalling receiver according to the present patent application, this is implemented by the application of discrete Fourier transformation (DFT), calculated in each case for a value k_i given by $k_i/N = f_i/f_S = f_i \times T_s$,

where

k_i is a rank number indicating one of the nominal signalling frequencies,

f_i is the relevant nominal signalling frequency,

N is the number of samples over which the DFT is calculated,

f_S is the sampling frequency, and

T_s is the sampling interval.

For a DFT it may benerally be written:

$$F(k) = \sum_{n=0}^{N-1} f(n,T_s) \left\{ \cos(2\pi f,T_s)n - j\sin(2\pi f,T_s)n \right\}$$

where

n is the rank number of a sample of the series of N samples, and $f(n,T_s)$ is the n^{th} sample of this series.

It is generally known, for example from the

above-quoted articles by Harris, that a result of a calculation over a series of N samples can be improved in DFT by using so-called Kaiser-Bessel window functions. A function of this type is generally given by:

$$W(n) = \frac{I_o\left[\pi.a(1,0-(n/0.5\ N)^2)^{\frac{1}{2}}\right]}{I_o(\pi.a)}$$

where

$$I_o(x) = \sum_{1=0}^{\infty} \left[(x/2)\ ^1/1!\right]^2$$

represents the so-called modified zero-order Bessel function. The properties of the window function are in the main determined by the choice of the parameters a and N. A frequency response is obtained by determining the modulus $|F(k)|$ as the root of the sum of the squares of the two parts

$$\sum_{n=0}^{N-1} W(n)\ \cos\ (2\pi nk/N) \text{ and } \sum_{n=0}^{N-1} W(n)\ \sin\ (2\pi\ nk/N).$$

It appears that the "height" of the side lobes belonging to a main lobe, and also the width of this main lobe are chiefly determined by the choice of the parameter a, while the "height" (the maximum or top) of the main lobe is dependent on the magnitude of the parameter N. It is customary in describing the filter characteristics to use a frequency (bin) normalized on the centre frequency (nominal frequency) of a main lobe. A bin is defined as a fundamental frequency interval, where $f_S/N = 1/NT$ Hz.

In a touch-tone signalling system the nominal signalling frequencies that are divided over the two different frequency bands, i.e. the low-band subgroup and the high-band subgroup, typically occur with different mutual differences in the frequency spectrum. The bandwidths of the different detectors are also mutually different. Further, for detection in a touch-tone signalling system it is necessary to take account of the possible presence of a strong call tone (nominal 150 and

450 Hz). With a TDK telephone set it is usual to switch the microphone off when a key on the keyboard is pressed and a combination of two signalling frequencies is sent out. Apart from this precautionary measure, a TDK signalling receiver should be equipped to guard against the influence of speech and backgroundnoise. Speech and background noise can reach the input of a signalling receiver during the phase in which a connection is being built up. Such speech and background noise have the following possible sources:

1) previous to the dialling (selection) of the first digit and between the selection of successive digits, the microphone and the subscriber's set is connected with the subscriber line; and

2) during the time that the signalling receiver is connected, crosstalk on the subscriber lines can also cause interfering signals to reach the input of this signalling receiver. Such interfering signals, such as speech or background noise, can disturb the proper operation of the signalling receiver through the crossover of calling tones. On the one hand, such interfering signals may contain components which, in frequency, amplitude and duration, may be interpreted as valid, and on the other hand such signals may lead to the rejection of a valid signal that is present.

The spurious-signal guard function, also referred to as speech guard function, is fulfilled in a signalling receiver according to the present invention by provisions which, on the basis of the results of the eight digital signal-processing devices, determine the two largest of these results and add together the energy contents of the other six of these devices and then comparing the sum result with a threshold value which is derived from the two largest detected results. It is advantageous for this purpose to use a device which makes available the sum result of the squared results of these digital signal processing devices other than those that possess

the largest energy content. It has been found that for the detection of one of the nominal signalling frequencies in the low-band subgroup and in the high-band subgroup, respectively, a number N of respectively 256 and 144 samples of the input signal must be taken and that the lowest value that can be used for the window parameter a is a = 2. In connection with the speech guard function, it is necessary to analyse the largest possible frequency band. To this end, however, each detector should introduce a high attenuation for a neighbouring signalling frequency of the same subgroup. The smallest value for the minimum frequency distance Δf between a nominal signalling frequency and a frequency that limits the bandwidth of the neighbouring detector applied to the lowest frequencies from the low-band subgroup, i.e. the frequencies 697 and 770 Hz. Assuming that the half bandwidth of each of the detectors is given as 1.5 % (of the relevant nominal signalling frequency) plus 2 Hz, the said minimum frequency distance Δf of the detector for the nominal signalling frequency is 697 Hz when Δf = 59.5 Hz. As illustrated in fig. 5, tables have been calculated from which, for different values of the window parameter a, the attenuation can be read for a detector at a given value of the above-mentioned minimum frequency distance Δf expressed in bins. It follows from fig. 5, for example, that for a minimum attenuation value of 30 dB to be introduced for a nominal neighbouring frequency, for a value a = 2, the minimum frequency distance Δf must be about 2 bin. The number N of samples that has to be taken of the relevant signal is then given by $N = T_S \times 2/\Delta f = 0.000125 \times 2/59.5 = 256$. This assumes a sampling frequency f_S of 8000 Hz or a sampling interval T_S of 125 μs. Choosing the nearest binary number, it then follows that the number of samples is N = 256. In a similar manner it follows for the high-band subgroup that the smallest value for Δf is approximately 105 Hz. In this connection the DFT for the detectors of this high-

band should be carried out over a number N of 144 samples (144 = 256 x 697/1209).

Assuming that a digital signal processing device is available that is capable of repeatedly carrying out a DFT over 128 samples within a frame time interval of 125 μs, there should be a provision that will make it possible, even in a situation where 256 samples have to be taken of an input signal, for calculating a DFT over this number. For this purpose, according to one aspect of the invention, the signal receiver is equipped for reading out from the input memory 3 for each of the last 256 samples of the eight channels stored, only the even or only the odd numbered samples, so that in fact only 128 or 72 samples, respectively, are involved in the DFT operations. This negation of half the number of available samples amounts in fact to halving the sampling frequency from 8000 Hz to 4000 Hz. The form of a frequency response characteristic, also referred to as the detector response, is not affected by this; however, the response over the frequency band from 0-2000 Hz is mirrored with respect to 2000 Hz. This means, for example, that a detector which responds to a frequency f_1 gives the same result for a frequency $4000-f_1$. Such mirroring is advantageous for the speech guard function since the analysed frequency range is thereby doubled.

As appears from fig. 5, the digital signal guard device designed for the nominal signalling frequency of 697 Hz introduces an attenuation of 27.5 dB at the values N = 256 and a = 2 for a minimum frequency distance Δf = 59.5 Hz. The maximum attenuation in the passband is thereby smaller than 1 dB. According to a further aspect of the invention, for each of the digital signal processing devices designed for the nominal signalling frequencies of the low-band subgroup the window parameter a is chosen such that all these devices at a frequency distance $b_1 = \Delta f$ introduce an attenuation d_1 which in principle has the same value. This is an advantageous feature for the

speech guard. The foregoing is summarized below in table 1.

TABLE 1

f_{Nom}	a	b1(Hz)	bb1(bin)	b2(Hz)	bb2(bin)	d1(dB)	d2(dB)
697	2.00	59.45	1.90	12.46	0.40	27.6	0.93
770	2.10	60.55	1.94	13.55	0.43	27.1	1.03
852	2.60	68.45	2.19	14.78	0.47	27.3	1.03
940	3.00	74.22	2.37	16.12	0.51	27.4	1.06

In this Table:

f_{Nom} is the nominal signalling frequency

a = parameter of the Kaiser-Bessel window function

b_1 = minimum frequency distance Δf between relevant nominal signalling frequency and a frequency that limits the bandwidth of the neighbouring detector

bb1 = the same as b_1 but now expressed in bins ($bb_1 = b_1 \times N \times T$)

b_2 = detected bandwidth in Hz (1.5% nominal frequency +2 Hz).

bb_2 = the same as b_2 but now expressed in bins

d_1 = attenuation at the distance bb_1

d_2 = attenuation at the distance bb_2.

In Table 2 below the above-mentioned values are summarized for the DFT devices of the high-band sub-group.

TABLE 2

f_{Nom}	a	b1(Hz)	bb1(bin)	b2(Hz)	bb2(bin)	d1(dB)	d2(dB)
1209	2.00	104.96	1.89	20.14	0.36	27.2	0.75
1336	2.05	106.87	1.92	22.04	0.40	27.3	0.91
1477	2.50	118.96	2.14	24.16	0.43	27.3	0.88
1633	3.00	131.85	2.37	26.50	0.48	27.4	0.94

According to an important aspect of the present invention, the digital processing devices of the signalling receiver are implemented in such a way that two side lobes are added to each main lobe whose maximum is centred on one of the relevant nominal signalling frequencies, the

side lobes added to the main lobes for the low-band subgroup and high-band subgroup, respectively being situated respectively below and above the frequency band occupied respectively by the low-band subgroup and the high-band subgroup. Thus, to the eight detectors for the eight nominal signalling frequencies there are in fact four detectors added which operate at the limits of the low-band subgroup and the high-band subgroup. The results of these four detectors are then comprised in the speech guard described in the foregoing. By such an addition of four detectors the frequency spectrum analysed for the speech guard function is appreciably widened. A further widening is obtained by making the response of these four added detectors as wide as possible.

The requirements to be placed on these four added detectors are the following:

a. the detector operating at the underside of the low-band should introduce a relatively high attenuation for calling tones (maximum 470 Hz);

b. the detector operating at the upper side of the low-band should introduce a high attenuation for the frequency of 1188.8 Hz that forms the lower limit of the detection bandwidth of the digital signal processing device for the lowest nominal signalling frequency (1209 Hz) of the high-band;

c. the detector operating at the underside of the high-band should introduce high attenuation for the frequency that forms the upper limit of the detection bandwidth of the digital signal processing device for the highest nominal signalling frequency (941 Hz) of the low-band;

d. the detector operating at the upper side of the high-band should introduce a high attenuation for the mirror frequency of 2000 Hz; and

e. for all four of these added detectors the attenuation for the neighbouring frequency should be greater than 27 dB.

It has been found that for these four added

detectors the window parameter a has its maximum useful value value at a = 3.6. Extrapolating from the graph in fig. 5 results in that at a = 3.6 and a number of samples N = 256 an attenuation of 27.5 dB is introduced at a frequency distance of 2.62 bin. It follows from this that the minimum frequency distance Δf between the nominal frequency of the relevant detector that has to be added to the low-band and the frequency that forms a limit for the detection bandwidth of the detector for the neighbouring nominal signalling frequency is given by

$\Delta f = (\text{bins} \times f_S)/N = (2.62 \times 8000)/256 = 81.87$ Hz. The lowest limit frequency of the low-band is given by 697 - 12.46 = 684.5 Hz. The nominal frequency of the extra detector that must be added to the underside of the low-band is thus given by 684, -81.87 $\cong$ 603 Hz. The attenuation introduced at the frequency of 470 Hz is here $>$ 90 dB.

The highest limit frequency of the low-band is given when

941 + 16.12 = 957.1 Hz. The nominal frequency of the added detector that must operate at the upper side of the low-band is then given when 957.1 + 81.87 $\cong$ 1039 Hz. The attenuation introduced at the frequency 1188.8 Hz is then $>$ 90 dB.

The lowest limit frequency of the high-band is given when 1209 - 20.14 = 1188.8 Hz. The extra detector at the underside of the high-band then has a nominal frequency given by 1188.8 - 81.87 = 1107 Hz. The attenuation for the frequency 957.1 Hz is then $>$ 90 dB.

The added detector for the upper side of the high-band must then be dimensioned for a number of samples N = 144. From this there follows a value for the minimum frequency distance Δf given as $\Delta f = (2.62 \times 8000)/144 =$ 145.5 Hz. The nominal frequency of this added detector for operation at the upper side of the high-band is then given as 1659.5 + 145.5 = 1805 Hz. The attenuation at 2000 Hz is then $>$63 dB.

Fig. 3 illustrates the frequency response of the digital signal processing devices for the nominal signalling frequency of 770 Hz and 1336 Hz, respectively, on the main lobe to which the two side lobes have been added for the four added detectors described in the foregoing.

Fig. 4 illustrates the speech detection characteristic formed by the frequency responses and which is operative with a signalling code formed from the nominal signalling frequencies of 852 and 1336 Hz, and whereby an extra filter has been added at the underside and upper side of the low-band and high-band respectively. Fig. 4 also illustrates that the points where neighbouring lobes of the speech detection characteristic intersect each other lie at practically the same level. The window functions chosen for this purpose can correct possible variations in the heights of the lobes mutually, including those of the side lobes added at the limits of the low-band and the high-band. A point of intersection at the aforesaid level is also given by the side lobe added to the upper side of the low-band and the side lobe added at the lower side of the high-band.

The block diagram of the signalling receiver given in fig. 1 illustrates a physical components structure for implementing the functions described in the foregoing.

In this diagram the box 3 denotes a freely accessible input memory with a storage capacity of 256 samples for each of the eight channels. The input and output of this input memory are regulated from a control device 4. Thiscontrol device comprises a part 5 for reading out the input memory in such a way that, for example, only the odd numbered samples are read out from a series of 256 channel samples. This constitutes in fact a halving of the number of signal samples read in per observation window of 32 ms (256 x 125 μs), which means that frequency components above 2 kHz, also have an influence. Since the response is mirrored relative to 2 kHz, frequency components

between 2300 and 2850 Hz and 3020 and 3340 Hz will contribute to the result of the speech detector.

The samples read out from the input memory are representative of PCM input signals. For the implementation of the DFT's it is recommendable to have the input signals available in linear form. To this end the signalling receiver contains a converter 6 which is equipped to linearize the input signals fed into it. At theoutput of the converter 6 a linearized input signals a_i appears with a range of values from -2047 - +2047. In this way only 12 of the 16 available bit positions are used. The 12bit input samples thus linearized are fed to the digital signal processing devices, going respectively to the inputs of two product accumulators of said devices 7 and 8. Fed to the other inputs of these product accumulators are 8-bit coefficients b_i and c_i respectively. The coefficients required for implementing the DFT's are precalculated and stored in two coefficient memories 9 and 10. In total these are required to store 4 x 2 x 128 words of 8 bits for the low-band and 4 x 2 x 72 words of 8 bits for the high-band. For simplicity, however, 4 x 2 x 128 words are also stored for the high-band, for which purpose the 4 x 2 x 72 words are supplemented with zeros. To obtain the frequency response characteristics, or speech detection characteristics as they are also called, of the form shown in Fig. 4, these coefficients are given as the product of the relevant sine and cosine coefficients, respectively and the corresponding window coefficients, in accordance with the following formulae:

$$b(i) = W_1(n,T)\sin(2\pi fTn) + \frac{1}{\sqrt{3}} W_2(n,T)\sin(2\pi .603.t.n) + \frac{1}{\sqrt{3}} W_3(n,T)\sin(2\pi .1039T.n)$$

$$c(i) = W_1(n,T)\cos(2\pi fTn) + \frac{1}{\sqrt{3}} W_2(n,T)\cos(2\pi .603.T.n.) + \frac{1}{\sqrt{3}} W_3(n,T)\cos(2\pi .1039\ T.n)$$

for the low-band

$$b'(i) = W_1'(n,T)\sin(2\pi fT.n) + \frac{1}{\sqrt{3}} W_2'(n,T)\text{sind}(2\pi.107T.n) + \frac{1}{\sqrt{3}} W_3'(n,T)\sin(2\pi.1805.T.n)$$

$$c'(i) = W_1'(n,T)\cos(2\pi fT.n) + \frac{1}{\sqrt{3}} W_2'(n,T)\cos(2\pi.1107.T.n) + \frac{1}{\sqrt{3}} W'_3(n,T)\cos(2\pi.1805.T.n)$$

for the high-band. In other words, for each nominal signalling frequency it is necessary to store 128 words for the sine terms and 128 words for the cosine terms. Chosen for this are the Kaiser-Bessel window functions W1, W2, W3; W'_1, W'_2, W'_3 etc. in accordance with the descriptions given in the foregoing. Each of the product accumulators 7, 8 operates under the control of the control device 4 in determining the DFT for the relevant nominal signalling frequency from the presented series of 128 signal samples b(i) and c(i) of the coefficient memories 9 and 10, respectively, in accordance with

$$x = \sum_{i=0}^{127} a(i) * b(i)$$

for the product accumulator 7, and in accordance with

$$y = \sum_{i=0}^{127} a(i) * c(i)$$

for the product accumulator 8. Here the x term and the y term are representative of the imaginary and real parts respectively of the relevant DFT, the filter result for the relevant nominal frequency then being given as the modulus of this DFT, which modulus in this exemplary embodiment is determined as the root of the sum of the squares of these real and imaginary parts of the relevant DFT. For this purpose an available 8-bit microprocessor 11 can be used, such as for example the Z-80. The operation can be split into the following two parts:

1. Conversion of the parts of the DFT available in linear form and the outputs of the product accumulators into a

logarithmic 8-bit wide form. Such a conversion offers the advantage that fewer bits are then necessary so that the actual calculations are simplified.

2. Application of an operation whereby the root of the sum of the squares is determined. In order that the 24-bit-wide numbers that are available at the outputs of the product accumulators can be converted into logarithmic form, so that a microprocessor of the aforesaid type can perform the further operations, a conversion known in the art as piecewise linear approximation can be carried out. In forming the real and imaginary parts of the relevant DFT it is permissible, before causing the said logarithmic conversion to take place, to eliminate the seven least significant bits from the presented signal samples as well as the corresponding character bit, so that for the following operations there are available 16-bit-wide numbers from which the logarithmic form can be determined. It is noted here that the product coefficients b_i and c_i, respectively, and more in particular the products of the sine and cosine terms and the relevant window function, are truncated to 8-bit-wide words in two's complement form. Proceeding from the 16-bit-wide numbers available in linear form in the product accumulators 7 and 8, the modulus m of the relevant DFT is given by

$$m = 16.{}^2\log \frac{\sqrt{x^2 + y^2}}{8}$$

Such an operation can be performed by the processor which forms part of the digital signal processing device and which may also form part of the microprocessor indicated in its generality by block 11.

The two product accumulators 7 and 8 are capable of processing the series of 128 signal samples and product coefficients presented to it into time intervals of 125 microseconds. This means that for calculating 8 DFT's or for eight nominal signalling frequencies, a time interval of 1 ms is required and for processing eight channels time

intervals of 8 ms are required. As remarked in the foregoing, the DFT's are calculated in overlapping form, which implies that each new DFT calculation is performed partly over the same samples as those that were involved in the previous DFT calculation, so that in every interval of 8 ms new information is obtained about the input signal. In each calculation, proceeding from 256 read-in signal samples, each time interval of 32 ms thus involves 24 ms of "old" signal and 8 ms of "new" signal. This overlapping detection is also controlled by the control device 4.

With the device described in the foregoing it is achieved that the results of every eight digital signal processing devices or detectors are available as 8-bit-wide logarithmic numbers which are stored in the memory of the microprocessor. From this, both for the low-band and for the high-band the largest of the relevant four results is determined, denoted respectively by IZL and IZH. The microprocessor is now further equipped for performing the three following determinations:

1. Determination of whether IZL or IZH, respectively, is greater than a threshold value L_1;
2. determination of whether the difference between the values IZL and IZH is smaller than a threshold value L_2; and
3. determination of whether the result of the speech detector is smaller than a threshold value L_3.

The result of the relevant speech detector is calculated from the aforesaid available results of the eight detectors, on the understanding that when it has been determined for the low-band and the high-band which detectors have produced the largest results, the result of the speech detector is determined by calculating from the other six detector results the root of the sum of the squares, so that these detectors are in fact combined into a speech detector with a broadband characteristic. The speech detector result is then obtained by adding together the 6-bit-wide numbers produced by the processor 12 and which

are representative of the aforesaid other six detector results.

The threshold value L_1 and L_2 are determined on the basis of the following considerations. In the first place, the absolute levels of the signals in a TDK signalling system are established. These levels are defined relative to a selected level A. The maximum level of a TDK signal is given by

a) two components with a level of A + 25 dBm;

b) a call tone with a level of A + 22 dBm; and

c) spurious components with a level of A + 25-20 dBm.

In a PCM system the maximum level should be lower than 0 dBm. In this connection, A = -34 dBm has been chosen, so that the TDK frequency response components have an absolute level between -34 and -0 dBm. The call tone has a maximum level of -12 dBm. By now normalizing the detector results with respect to the maximum detector result, the following values can be established for the thresholds L_1 and L_2:

$L_1 = 80 \times - 0.376 = -30.08$ dB and

$L_2 = 25 \times -0.376 = - 9.40$ dB.

The value of the threshold L_3 is established by determining the maximum result of the speech detector in the presence of a valid combination of nominal signalling frequencies. Such a maximum result arises when: two nominal signalling frequencies are present with a maximum frequency deviation such that these signals come through maximally via two neighbouring detectors (attenuation -27.1 dB), and spurious components exist with a level of 20 dB below the low-band component with a minimum level of -50 dB. Taking account of the imposed CEPT standards, this noise threshold L_3 is fixed at -17.25 dB below the level of the largest result IZL of the low-band, with a minimum of -42 dBm. When a result of the speech detector described in the foregoing turns out to be larger than this noise threshold L_3, the presented channel signal from which this speech detector result is derived is then rejected.

When the speech detector result is smaller than this noise threshold and the detector results IZL and IZH satisfy the requirements specified under 1) and 2) in the foregoing, it must be established whether the so-called character recognition condition exists. If this condition does exist, there appears at the output of the signalling receiver the correct signalling code. In this connection the signalling receiver is equipped for implementing a routine by means of which the imposed time requirements are satisfied by whether or not the so-called signal condition follows, that is the state at the input of the signalling receiver that exists when a signal at its input corresponds to a valid signal accompanied by an acceptable amount of unwanted frequencies, so that a decision can be made as to the validity of the signal. As described in the foregoing, a decision is made every 8 ms by the described signal condition operations as to whether or not one of eight signal conditions exists. To implement the aforesaid routine, which must be run in aconnection with the time requirements imposed, the signalling receiver is further designed such that a channel signal of short duration (< 20 ms) results at the most in a decision indicating that a signal condition exists. In order for a character recognition condition to arise it is therefore required that the same signal condition be present at least twice in succession. A signal interruption (< 20 ms) has the result that at the most five times in succession a decision is taken indicating that the signal condition does not exist.

The decision that the character recognition condition does not exist may thus only be taken when six times in succession a different signal condition arises than that which gave rise to the existence of the character recognition condition. The occurrence twice in succession of the same signal condition results in this case too in a decision that a character recognition condition exists. When a routine as described above has led to the decision

that a character recognition condition is present, an output message is composed which is sent out for further processing via the output unit indicated by 13 in fig. 1.

In the example of an embodiment of a signalling receiver according to the invention as described in the foregoing, a frequency spectrum of approx. 550 Hz to 1900 Hz is analysed for the purposes of the speech guard by the addition of said side lobes to each of the frequency responses of the digital signal processing devices. Such a speech detection response also occurs in mirrored form arouns 2 kHz when in the manner described a sampling frequency of 4 kHz is applied. In this way the selectivity of the devices is also appreciably improved. The aforesaid can be achieved with a physical components structure similar to that of a signalling receiver as described in aforesaid Netherlands patent application, without said component structure having to be expanded.

It is consequently possible in principle to combine receiving equipments for TDK signalling on the one hand and for MFC signalling on the other into a unit wherein each of the relevant receivers operates either as a TDK receiver or as an MFC receiver. For both types of receiver mainly the same physical components and the same routines incorporated therein can be used. The coefficients for both the TDK and the MFC receiver are stored and held available in the coefficient memories (PROM). The microprocessor 11 is correspondingly equipped for implementing the various test routines that are needed for both types of receiver.

CLAIMS

1. Receiving equipment which is arranged to recognize out of incoming signals a number of different frequency combinations, each composed of at least two different combinations of a number (s) of nominal voice-frequency signalling components, said equipment containing a number (s) of mutually different DFT digital signal-processing devices each having a frequency response characteristic the top of whose main lobe corresponds to one of said nominal voice-frequency signalling components situated within a frequency band as determined by one of two subgroups into which said voice-frequency signalling components are divided, and a processor which is equipped to process the results of the digital signal-processing devices, to detect two nominal voice-frequency signalling components that are received with the greatest strength, and to evaluate the other results so as to fulfil a voice-signal guard function, characterized in that the frequency response characteristic of each of the DFT digital signal-processing devices has a first side lobe and a second side lobe, each with one top, wherein the top of the first side lobe and the top of the second side lobe are situated respectively below and above the frequency band of the subgroup to which the relevant nominal signalling frequency component belongs; and the processor is equipped a) for repeatedly adding together the results of a number (s-2) of the DFT devices in such a way that a sum result is obtained that is representative of the total energy content of all detected input signals other than two nominal voice-frequency signalling components that are received with the greatest strength, and b) for comparing the sum result with a threshold value which is derived from one of said two nominal voice-frequency signalling components

that are received with the greatest strength.

2. Receiving equipment as claimed in Claim 1, characterized in that each of the digital signal-processing devices is dimensioned in such a way that at the minimum frequency distance between a relevant nominal signalling frequency component and a frequency thatlimits the detection bandwidth of the nearest neighbouring nominal signalling frequency component an attenuation is introduced which in the main has one and the same magnitude.

3. Receiving equipment as claimed in Claims 1 or 2, characterized in that each of the digital signal-processing devices is dimensioned such that the tops of the relevant two side lobes are mainly 6 dB lower than the top of the corresponding main lobe.

4. Receiving equipment as claimed in any one of the foregoing Claims 1-3, characterized in that the shape of the frequency response characteristic of each of the digital processing devices is determined by a window function of the kind known as the Kaiser-Bessel function.

5. Receiving equipment as claimed in Claim 4, characterized in that the digital signal-processing devices are so dimensioned that the frequencies of the tops of the side lobes that are added to the subgroup with the lowest and highest frequency band, respectively, are located at respectively 603 and 1039 Hz and 1107 and 1805 Hz.

6. Receiving equipment as claimed in any one of the foregoing Claims 1-3, comprising a memory device for the temporary storage of a number of samples of a presented signal, characterized in that read-out means are provided for alternately reading out half the number of samples stored in the memory device, provided that upon each read-out the most neighbouring sample of a read-out sample is negated.

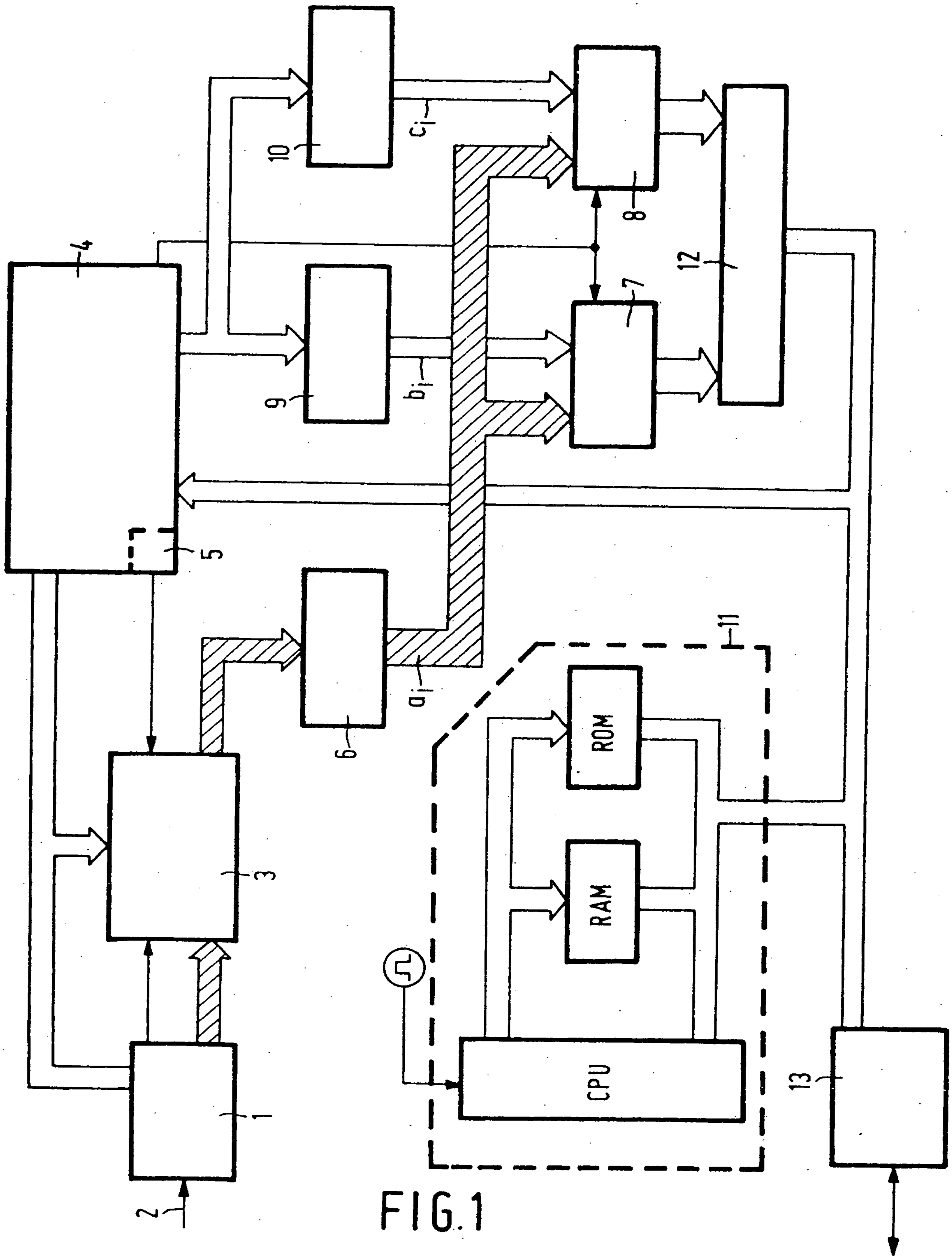


FIG.1

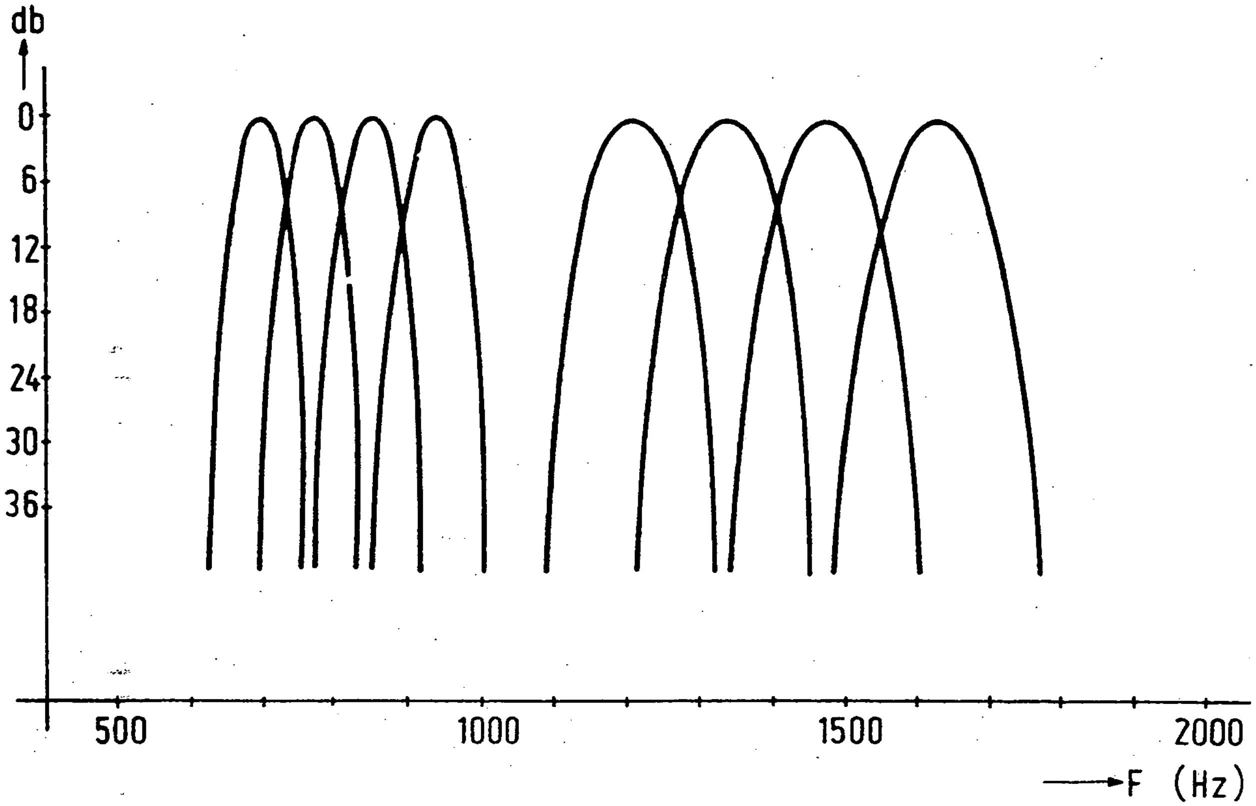

db
0
6
12
18
24
30
36
500
1000
1500
2000
F (Hz)

FIG.2

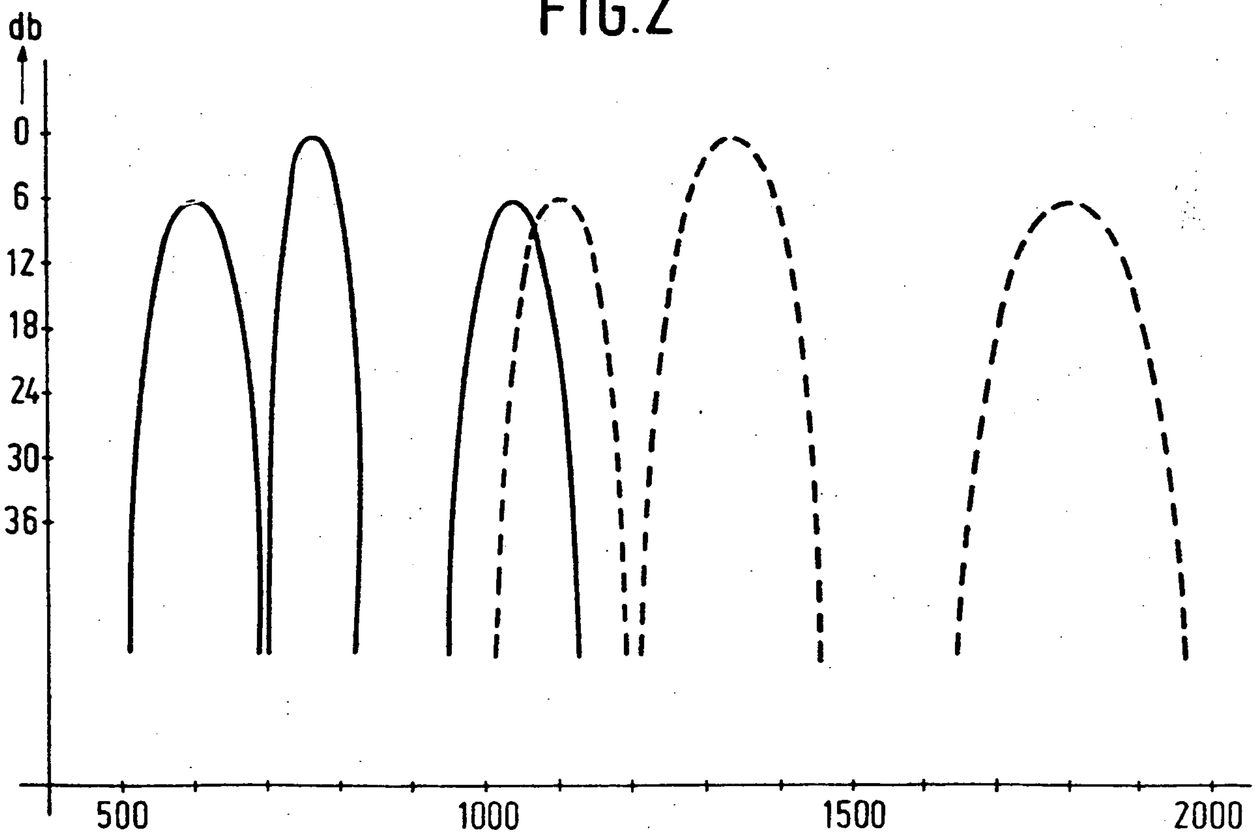

db
0
6
12
18
24
30
36
500
1000
1500
2000
F (Hz)

FIG.3

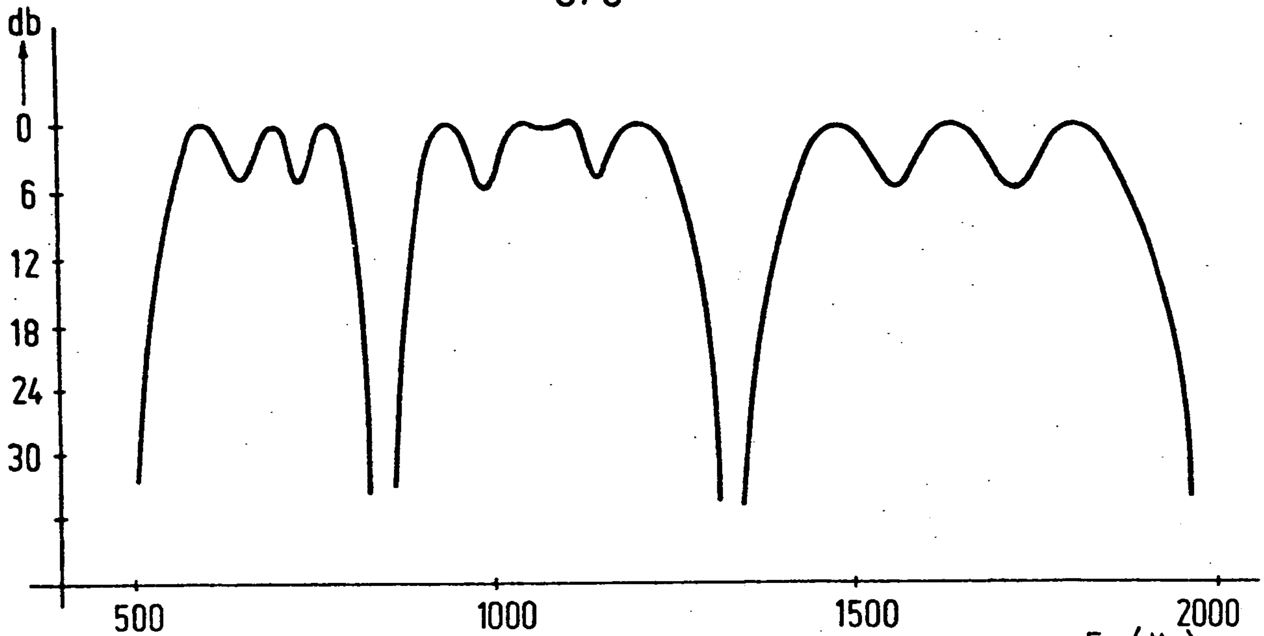

db
0
6
12
18
24
30
500
1000
1500
2000
F (Hz)

FIG.4

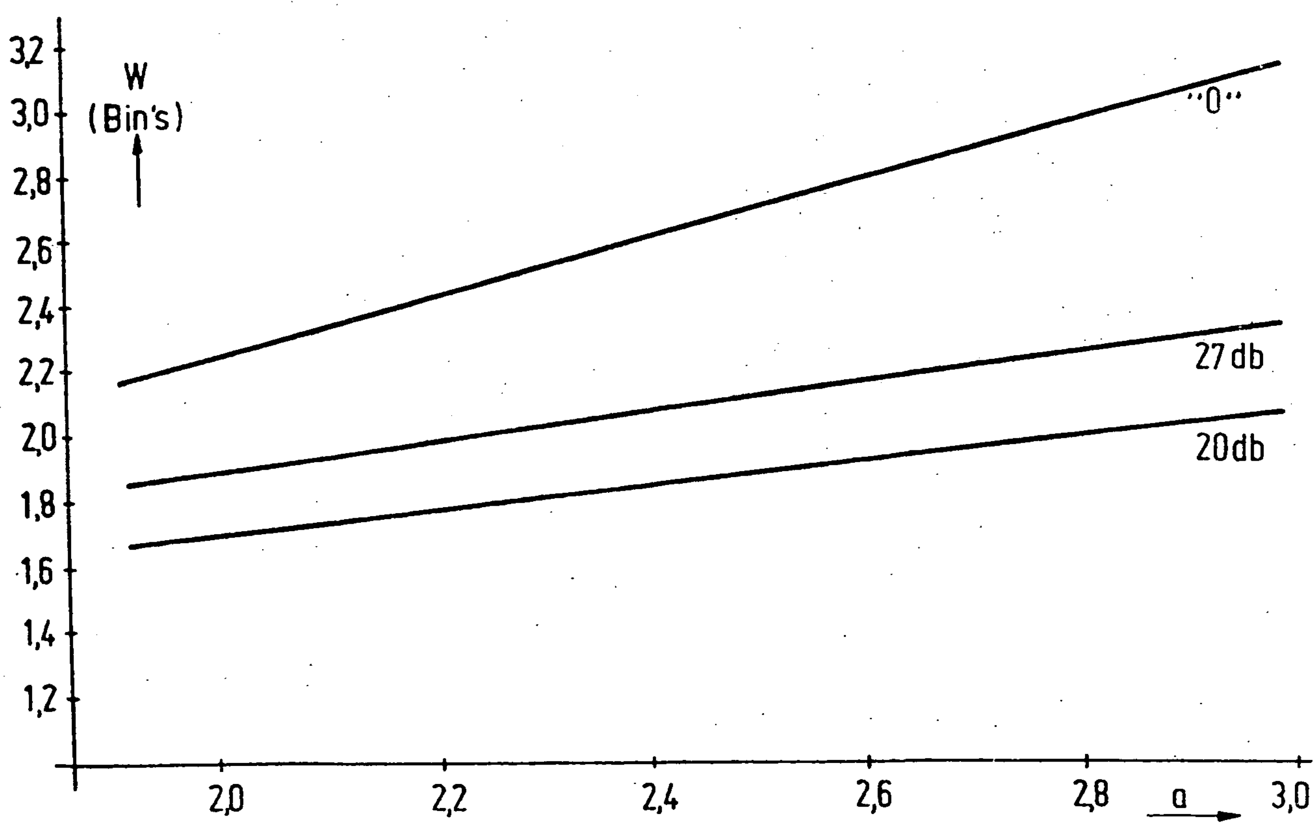

3,2
3,0
2,8
2,6
2,4
2,2
2,0
1,8
1,6
1,4
1,2
W
(Bin's)
"0"
27db
20db
2,0
2,2
2,4
2,6
2,8
3,0
a

FIG.5

European Patent Office

EUROPEAN SEARCH REPORT

0213651
Application number

EP 86 20 1201

DOCUMENTS CONSIDERED TO BE RELEVANT

Category	Citation of document with indication, where appropriate, of relevant passages	Relevant to claim	CLASSIFICATION OF THE APPLICATION (Int Cl 4)
Y	FR-A-2 455 405 (INTERNATIONAL STANDARD ELECTRIC) * Page 6, lines 28-37; page 15, line 11 - page 16, line 7; page 35, lines 1-20 * & US-A-4 355 405	1	H 04 Q 1/46
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Y	IEEE TRANSACTIONS ON COMMUNICATIONS, vol. COM-21, no. 12, December 1973, pages 1331-1335, New York, US; I. KOVAL et al.: "Digital MF receiver using discrete fourier transform" * Page 1333, left-hand column, line 25 - page 1335, left-hand column, line 6 *	1	
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A	IDEM	2,3	H 04 Q

A	IEEE INTERNATIONAL CONFERENCE ON COMMUNICATIONS, Denver, Colorado, US, 14th-18th June 1981, vol. 1, pages 11.5.1-11.5.5, IEEE, New York, US; J.P. HENRY et al.: "Digital multifrequency receiver using non recursive (FIR) filters designed with window functions" * Page 11.5.2, right-hand column, lines 24-31 *	1,4	
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The present search report has been drawn up for all claims

Place of search	Date of completion of the search	Examiner
THE HAGUE	15-10-1986	VERSLYPE J.P.

CATEGORY OF CITED DOCUMENTS

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L : document cited for other reasons

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0213651

Application number

EUROPEAN SEARCH REPORT

EP 86 20 1201

DOCUMENTS CONSIDERED TO BE RELEVANT

Category	Citation of document with indication, where appropriate, of relevant passages	Relevant to claim	CLASSIFICATION OF THE APPLICATION (Int Cl 4)
A	IEEE INTERNATIONAL CONFERENCE ON COMMUNICATIONS, Denver, Colorado, US, 14th-18th June 1981, vol. 1, pages 11.6.1-11.6.5, IEEE, New York, US; V.K. JAIN: "Phase-assisted IPDFT digital DTMF receiver" * Page 11.6.2, left-hand column, line 27 - right-hand column, line 36; page 11.6.4, left-hand column, line 9 - page 11.6.5, left-hand column, line 15 *	1	
A	ERICSSON REVIEW, vol. 59, no. 1, 1982, pages 14-21, Stockholm, SE; J. REINIUS et al.: "Digital signal processing in system MD 110" * Page 18, right-hand column, line 33 - page 20, left-hand column, line 21 *	1,5	
A	GB-A-2 113 880 (PHILIPS) * Page 2, line 41 - page 3, line 33 *	1	TECHNICAL FIELDS SEARCHED (Int Cl 4)

The present search report has been drawn up for all claims

Place of search	Date of completion of the search	Examiner
THE HAGUE	15-10-1986	VERSLYPE J.P.

CATEGORY OF CITED DOCUMENTS

X particularly relevant if taken alone
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A technological background
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P intermediate document

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D : document cited in the application
L : document cited for other reasons

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